XDAIS compatible audio processing modules

eXpressDSP™ Compliant
DSP
TEXAS INSTRUMENTS

What is XDAIS?

XDAIS is an acronym for eXpressDSP Algorithm Interoperability Standard. DSP algorithms following this software standard work on all TI platforms and are dynamically interchangeable at runtime.
The algorithms described here are XDAIS compliant and have been tested and certified as such by Texas Instruments.

Compressor

The purpose of the compressor/limiter module is to „rein in" excessive signal dynamics of the incoming audio signal. This is done by reducing the gain in the audio path when the signal level exceeds a predetermined threshold. While the static characteristics of the compressor/limiter specify the system gain for different static loudness levels, the dynamic characteristics control the smooth reaction of the compressor to changed signal dynamics.

The static characteristics of the Compressor are determined by the following three parameters:

Threshold

This is the signal level above which the compressor starts to reduce the gain. If the signal level stays below this level the compressor will work as a simple bypass.

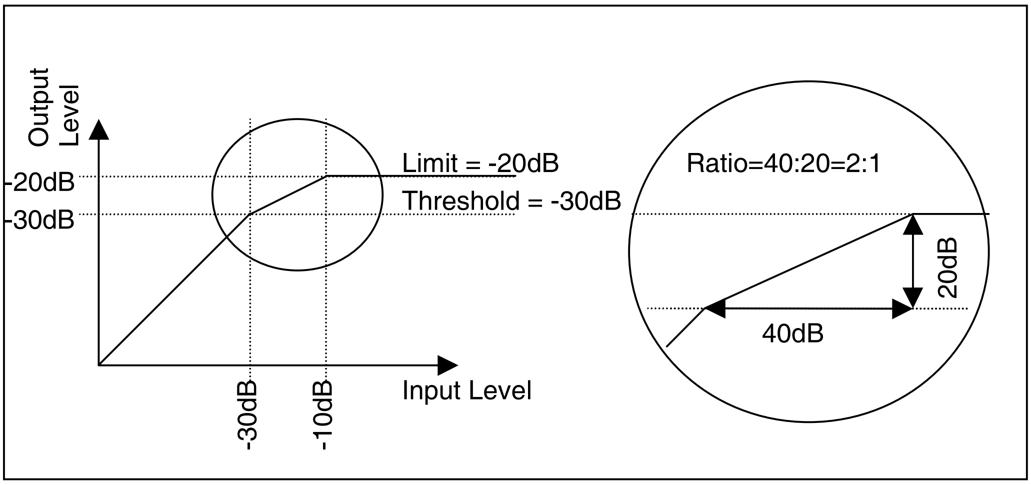
Ratio

This is a measure of how much the gain will be reduced if the signal level exceeds the threshold. A ratio of 2:1 means the signal is allowed to increase by only 1dB for every 2dB of input signal increase above the threshold. A ratio of 1:1 is equivalent to a bypass function.

Limit

This measure is the maximum possible output signal level. Exceeding signal levels will be reduced (clipped) to this value.

The figure explains these parameters with a static system function of the compressor/limiter.



The dynamic characteristics of the module can be controlled by the following Parameters:

Attack Time

This measure determines the rate at which the system gain is smoothly reduced if an increase of the input signal dynamics occurs. If the compressor works in look ahead mode, the attack time is chosen automatically to prevent signal clipping.

Release Time

This measure determines the rate at which the system gain rises following a decreased signal dynamics.

Stereo

Stereo mode of the compressor/limiter can be switched off. In stereo mode the maximum of both signal levels is taken into account. The same gain is applied to both audio channels. In mono mode only one channel is processed.

Look Ahead Mode

It is possible to improve the reaction time to a signal attack by introducing a small delay into the signal path. The gain will be adjusted smoothly preventing a clipping of the signal attack.

5-Band Equalizer

The 5-band equalizer module processes one audio data channel and contains 5 filter modules with lowcut, low-shelf, 2 mid-band, and high-shelf characteristics. The controllable parameters of each filter type are explained in Figure 2. All parameters can be changed at runtime. The band number and filter configuration for each band can be easily modified by request.

Low Cut Filter

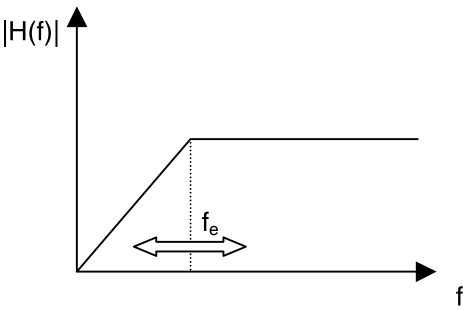
The low cut filter module allows an attenuation of spectral signal components below the given cut off frequency. It is useful to reduce room rumbling and microphone thumps. A useful cutoff frequency would be around 75 Hz.

Low Shelf / High Shelf Filter

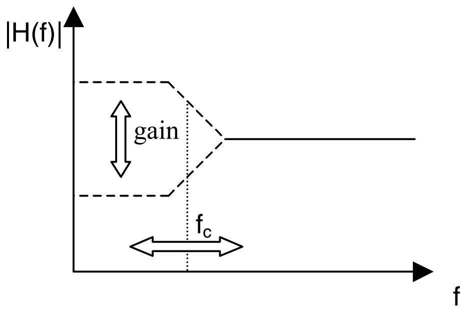
The Shelf filter modules boost or cut frequencies below or above the cutoff with an adjustable gain, and pass frequencies above or below the shelf cutoff with no change made to their gain. Use this effect to enhance or diminish any amount of low or high frequency material in a sound.

Parametric Mid Band Filter

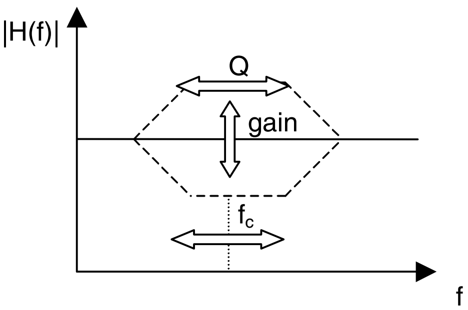
This is a very flexible filter, capable of creating notches or peaks without influencing nearby frequencies. Select a center frequency to process, the amount of gain change and the Q factor, which determines the bandwidth of the filter.



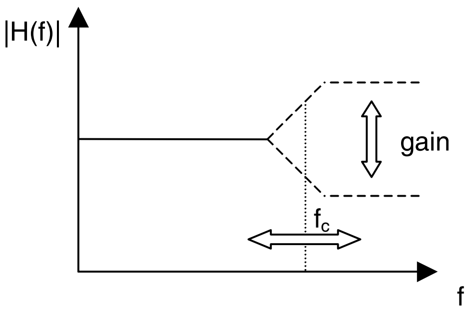
Low Cut Filter



Low Shelf Filter



Parametric Mid Band Filter

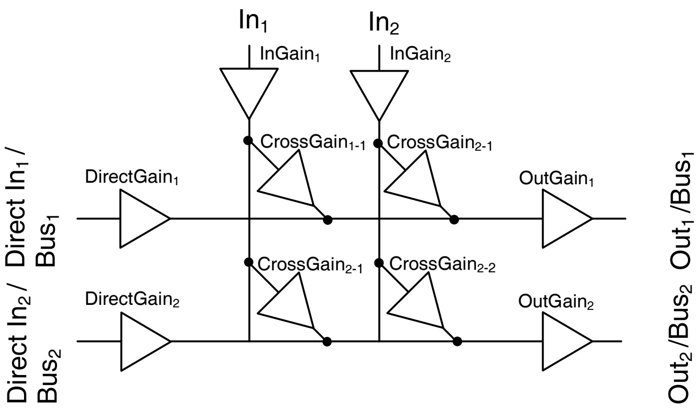


High Shelf Filter

NxM Mixer

The mixer module represents a NxM mixing matrix for audio applications. Basically, this is a type of mixer that allows any input to be routed to any output. N input channels are mixed with an M-channel direct input audio bus, which is mapped to M output channels. Due to the direct inputs, mixers can be cascaded in a multi-DSP audio system to increase the number of input channels.
The number of input and output channels as well as input/output gain for each channel and crosspoint is adjustable. Each crosspoint between an input channel and a direct input channel can work in master or slave mode whereby master crosspoints control the gain of connected slave crosspoints.

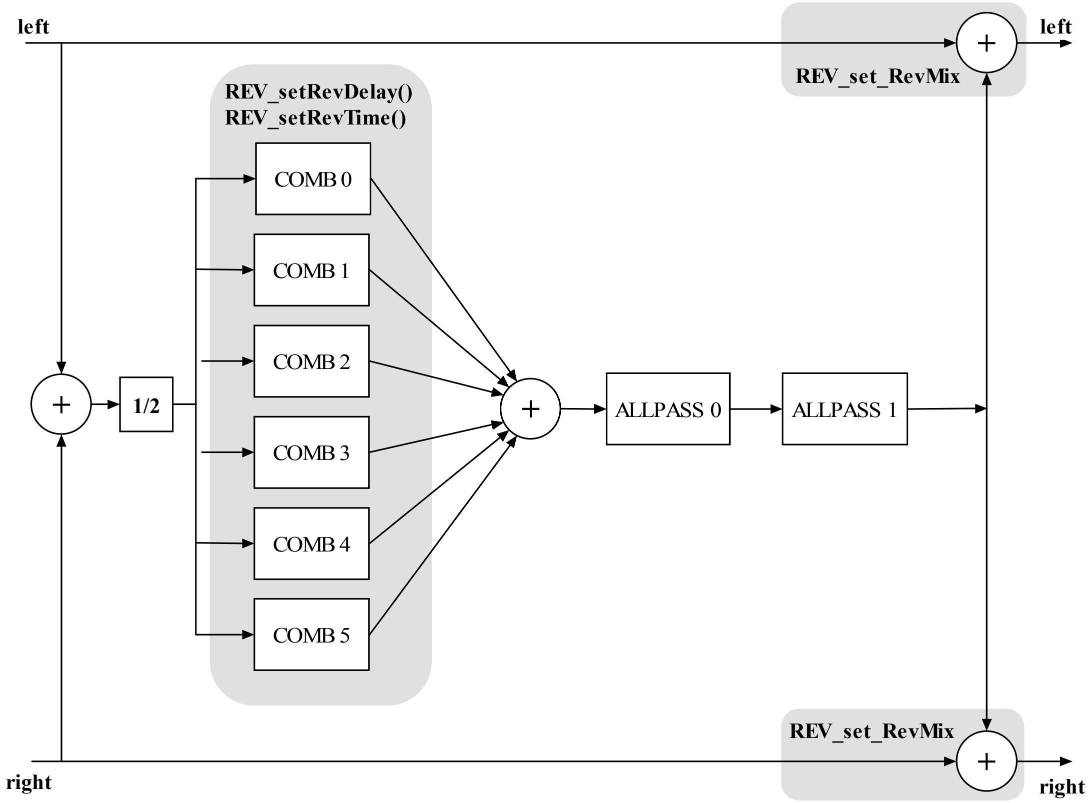
For a simple example of a 2x2 Mixer refer to the following figure.



Reverb

The reverb effect simulates the effect of sound reflections in a large concert hall or room. In dependency of the room size and the characteristics of the surrounding walls, the reflections decrease and fade to silences after a certain time.

The introduced reverb algorithm is based on the model of M. A. Schroeder and James A. Moorer (six parallel comb filters and two allpass filters in a cascade). The algorithm processes two channels (stereo signal), whereby the reverb signal is calculated out of the sum of both channels.



The following parameters can be used to change the characteristic of the reverberator:

Reverb Delay

This parameter defines the size of the room, given as the delay of the first reflection.
Possible values are 30 up to 100 ms.

Reverb Time

The reverb time controls the fade-out period of the algorithm.
This parameter can be set from 0 to 10 s.

Reverb Mix

This parameter controls the ratio (in percent) between the original input signal and the reverb signal in the output signal. Hence, the mix parameter controls the intensity of the reverb.